SCAD ENGINEERING COLLEGE

DEPARTMENT OF ECE

V SEMESTER ECE **QUESTION BANK**

EC6502 DIGITAL SIGNAL PROCESSING

UNIT I

DISCRETE FOURIER TRANSFORM

PART – A

1. Obtain the circular convolution of the following sequences $x(n) = \{1,2,1\}$; $h(n) = \{1, -2, 2\}$
2. How many multiplications and additions are required to compute N –point DFT using radix – 2 FFT?
3. Define DFT and IDFT?
4. State the advantages of FFT over DFTs?
5. What is meant by bit reversal?
6. Distinguish between DFT and DTFT?
7. What is zero padding? What are its uses?
8. Determine the number of multiplications required in the computation of 8 – point DFT using FFT?
9. What is twiddle factor?
10. How many stages of decimations are required in the case of a 64 point radix 2 DIT FFT algorithm?
11. Find the 4 – point DFT sequence $x(n) = \{1, 1, -1, -1\}$.
12. What is meant by in – place computation?
13. What are the differences and similarities between DIT and DIF.
14. Distinguish between linear convolution and circular convolution?
15. What are the differences between Overlap – add and Overlap – save method?
16. State the properties of DFT?

17. Draw the basic butterfly diagram for the computation in the decimation in frequency FFT algorithm and explain?
18. How will you perform linear convolution using circular convolution?
19. Find the circular convolution of *x(n)* = {1,2,3,4} with *h(n)* = {1,1,2,2}?
20. State Parseval's relation with respect to DFT?

PART - B

1. (i) Compute the eight point DFT of the sequence by using the DIF – FFT algorithm.

   1 0 7
   0

   (ii) Summarize the properties of DFT
2. Explain the Overlap add and Overlap save method
3. (i) Compute the eight point DFT of the sequence

$$\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0$$

   Using radix – 2 DIT algorithm.

   (ii) Explain Overlap add method for linear FIR filtering of a long sequence.
4. Explain Radix – 2 DIF FFT algorithm. Compare it with DIT – FFT algorithms.
5. Compute the linear convolution of finite duration sequences *h(n)* = {1,2} and *x(n)* = {1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1} by Overlap add method?
6. (i) Explain the following properties of DFT (a) Linearity (b) Complex conjugate property (c) Circular Convolution (d) Time Reversal

   (ii) Perform the linear convolution of the sequence x(n) = {1, -1, 1, -1} and h(n) = {1,2,3,4} using DFT method
7. (i)Derive the butterfly diagram of 8 point radix – 2 DIF FFT algorithm and fully label it.

   (ii) Compute the DFT of x(n) = {1, 1, 0, 0}
8. (i) Prove that FFT algorithms help in reducing the number of computations involved in DFT computation

   (ii) Compute a 8 point DFT of the sequence using DIT – FFT algorithm

$$1, 2, 3, 2, 1, 0$$

9. (i) Differentiate DFT from DTFT.

(ii) Compute an 8 point DFT of the sequence

$$1,\ 0,\ 1, -1,\ 1,\ 1, 0,\ 1$$

10. (i) Determine the N – point DFT of the following sequences

(a) x(n) = δ(n)   (b) x(n) = δ(n-1)

(ii)Compute 8 – point DFT of the sequence x(n) = {0, 1, 2, 3, 4,5, 6, 7} using radix – 2 DIF algorithm?

11. (i) Compute the 8 point DFT for the following sequences using DIT – FFT algorithm

$$\begin{matrix} 1 & -3 & 3 \\ 0 & & \end{matrix}$$

(ii) Summarize the Difference between overlap – save method and overlap – add method.

12. (i) Compute the DFT of the sequence whose values for one period is given by x(n) = {1,1,-2,-2}

(ii) Compute the eight point DFT of the sequence

$$\begin{matrix} 1 & 0 & 7 \\ 0 & & \end{matrix}$$

by using DIT algorithm

UNIT II

INFINITE IMPULSE RESPONSE FILTERS

PART – A

1. Compare Butterworth with Chebyshev filters?
2. What is known as pre warping in digital filters?
3. List the properties of Chebyshev filter?
4. Draw the direct form structure of IIR filter?
5. Why do we go for analog approximation to design a digital filter?
6. What is the advantage of direct form II realization when compared to direct form I realization?
7. Why the Butterworth response is called a maximally flat response?
8. Mention the advantages of cascade realization?
9. Convert the given analog transfer function H(s) = —— into digital by impulse invariant method?
10. Give the steps in design of a digital filter from analog filters?
11. What area the disadvantages of direct form realization?
12. What is frequency warping?
13. Compare IIR and FIR filters
14. Write the properties of Butterworth filter?
15. What is bilinear transformation?
16. What are the advantages and disadvantages of bilinear transformation?
17. Distinguish between recursive realization and non recursive realization?
18. What is meant by impulse invariance method of designing IIR filter?
19. Give the expression for location of poles of normalized Butterworth filter?
20. What are the parameters that can be obtained from Chebyshev filter specification?

PART – B

1. i) Explain the procedure for designing analog filters using the Chebyshev approximation.

   ii) Convert the following analog transfer function in to digital using impulse invariant mapping with T=1sec

   H(s)=————

2. i) Design a digital second order low pass Butterworth filter with cut off frequency 2200 Hz using bilinear transformation. Sampling rate is 8000 Hz.

   ii) Determine the cascade form and parallel form implementation of the system governed by the transfer function

   H(Z)=———

3. Design a digital Butterworth filter using impulse invariance method satisfying the constraints .Assume T=1sec

   0.8 | |) 1 ; 0 0.2

   | |) 0.2; 0.6

4. Obtain the direct form I ,direct form II and cascade form realization of the following system functions

   $$y(n)=0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)$$

5. Determine the system function H(z) of the chebyshevs low pass digital filter with the specifications

   1. =1dB ripple in the pass band 0 0.2
   2. =1dB ripple in the stop band 0. 3

   Using bilinear transformation (assume T=1sec)

6. Obtain the direct form I, direct form ii ,cascade, parallel form realization for the system

   $$y(n)= -0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)$$

7. Explain in detail Butterworth filter approximation

8. Explain the bilinear transform method of IIR filter design. What is wrapping effect? Explain the poles and zeros mapping procedure clearly.

9. i)Obtain the cascade form realization of the digital system

$$y(n)=3/4\ y(n-1)-(1/8)y(n-2)+1/3x(n-1)+x(n)$$

ii) Convert the given analog filter with transfer function H(s)=———— in to a digital IIR filter using bilinear transformation. Assume T=1sec.

10. Discuss the steps in the design of IIR filter using bilinear transformation for any one type of filter?

11. Apply Bilinear Transformation to H(s) =2/(S+2) (S+3) with T=0.1 sec.

12. Design a analog Butterworth filter that has a 2db pass band attenuation at a frequency of 20 r/sec & at least 10db stop band attenuation at 30 r/sec?

UNIT III

FINITE IMPULSE RESPONSE DIGITAL FILTERS

PART – A

1. Give the equations of Hamming window and Blackman Window. (Nov2010)
2. Determine the transversal structure of the system function

$$H(z) = 1 + 2Z^{-1} - 3Z^{-2} - 4Z^{-3} \qquad \text{(Nov2010)}$$

3. State the properties of FIR filter? (April2011)(Nov2013)
4. What is meant by Gibbs Phenomenon? (April2011) (May2012)
5. What are the desirable characteristics of window? (Nov2011) (Nov2013)
6. What are called symmetric and antisymmetric FIR filters? (May2012)
7. What are the features of FIR filter design using Kaiser's approach? (Nov2012)
8. Draw the direct form implementation of the FIR system having difference equation.

$$y(n) = x(n) - 2x(n-1) + 3x(n-2) - 10x(n-6) \quad \text{(Nov2012)}$$

9. What are the techniques of designing FIR filters?
10. What are the possible types of impulse response for linear phase FIR filters?
11. What is the principle of designing FIR filter using windows?
12. What is mean by FIR filter?
13. Write the steps involved in FIR filter design
14. What are advantages and disadvantages of FIR filter?
15. What is the reason that FIR filter is always stable?
16. State the condition for a digital filter to be causal and stable?
17. Compare Hamming window with Kaiser window.
18. What is the principle of designing FIR filter using frequency sampling method?
19. What is window and why it is necessary?
20. What is the necessary and sufficient condition for linear phase characteristic in FIR filter?

PART - B

1. Design a high pass filter with a frequency response

$$\omega \quad 1 \qquad \frac{}{2} \quad |\omega|$$
$$0 \qquad |\omega| \quad \frac{}{4}$$

Find the values of *h(n)* for N = 11 using hamming window. Find *H(z)* and determine the magnitude response.

2. (i) Determine the coefficients *{h(n)}* of a linear phase FIR filter of length M = 15 which has a symmetric unit sample response and a frequency response that satisfies the condition

$$\frac{2}{15} \qquad \begin{matrix} 1 & & 0,1,2,3 \\ 0 & & 4,5,6,7 \end{matrix}$$

(ii) Obtain the linear phase realization of the system function

$$\frac{1}{2} \quad \frac{1}{3} \qquad \frac{1}{4} \qquad \frac{1}{3} \qquad \frac{1}{2}$$

3. Realize the system function by linear phase FIR structure

$$\frac{2}{3} \quad 1 \quad \frac{2}{3}$$

4. Explain the designing of FIR filters using windows?

5. Design an ideal high pass filter using hanning window with a frequency response

$$\omega \quad 1 \qquad \frac{}{4} \quad |\omega|$$
$$0 \qquad |\omega| \quad \frac{}{4}$$

Assume N = 11.

6. Design a FIR low pass filter having the following specifications using Hanning window

$$\omega \quad \begin{matrix} 1 \\ 0 \end{matrix} \qquad -\frac{}{6} \quad |\omega| \quad \frac{}{6}$$

Assume N = 7

7. Design an FIR low pass digital filter using the frequency sampling method for the following specifications

Cut off frequency = 1500Hz

Sampling frequency = 15000Hz

Order of the filter N = 10

Filter Length required L = N+1 = 11

8. (i) Explain with neat sketches the implementation of FIR filters in direct form and Lattice form

   (ii) Design a digital FIR band pass filter with lower cut off frequency 2000Hz and upper cut off frequency 3200 Hz using Hamming window of length N = 7. Sampling rate is 10000Hz.

9. (i)Determine the frequency response of FIR filter defined by

$$y(n) = 0.25x(n) + x(n-1) + 0.25x(n-2)$$

   (ii) Discuss the design procedure of FIR filter using frequency sampling method.

10. Design an FIR filter using hanning window with the following specification

$$\omega \qquad 1 \qquad \overline{4} \quad \omega \quad \overline{4}$$
$$-2\omega \qquad \overline{4} \quad |\omega|$$

   Assume N = 5.

11. (i) Explain briefly how the zeros in FIR filter is located.

   (ii) Using a rectangular window technique, design a low pass filter with pass band gain of unity cut off frequency of 1000Hz and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7.

12. Consider an FIR lattice filter with coefficients $k_1 = 1/2$; $k_2 = 1/3$; $k_3 = 1/4$. Determine the FIR filter coefficients for the direct form structure

UNIT – IV

FINITE WORD LENGTH EFFECTS

PART – A

1. What is truncation? (Nov2010)
2. What is product quantization error? (Nov2010)
3. What is meant by fixed point arithmetic? Give example.(April2011)
4. Explain the meaning of limit cycle oscillator?(April2011)(Nov2012)
5. What is overflow oscillations? (Nov2011)(May2012)
6. What are the advantages of floating point arithmetic? (Nov2011)
7. Compare truncation with rounding errors. (May2012)
8. What is dead – band of a filter? (Nov2012)
9. What do you understand by input quantization error? (Nov2013)
10. State the methods used to prevent overflow? (Nov2013)
11. Compare fixed point and floating point arithmetic?
12. What are the two types of quantization employed in a digital system?
13. What is rounding and what is the range of rounding?
14. What is quantization step size?
15. Define Noise transfer function?
16. What are limit cycles?
17. What is meant by block floating point representation? What are its advantages?
18. What are the three-quantization errors to finite word length registers in digital filters?
19. What is coefficient quantization error? What is its effect?
20. Why rounding is preferred to truncation in realizing digital filter?

PART – B

1. Discuss in detail the errors resulting from rounding and truncation? (Nov2010)
2. Explain the limit cycle oscillations due to product round off and overflow errors? (Nov2010)
3. Explain the quantization process and errors introduced due to quantization? (May2011)
4. Explain how reduction of product round-off error is achieved in digital filters?(May2011)

5. Explain the effects of co efficient quantization in FIR filters? (May2011)
6. Distinguish between fixed point and floating point arithmetic
7. With respect to finite word length effects in digital filters, with examples discuss about
   (i) Over flow limit cycle oscillation
   (ii) Signal scaling
8. Consider a second order IIR filter with

$$\frac{1.0}{1 - 0.5 \qquad 1 - 0.45} \quad .$$

   Find the effect on quantization on pole locations of the given system function in direct form and in cascade form. Assume $b = 3$ bits.
9. What is called quantization noise? Derive the expression for quantization noise power.
10. How to prevent limit cycle oscillations? Explain.
11. (i) Explain the finite word length effects in FIR digital filters

    (ii) Represent the following numbers in floating point format with five bits for mantissa and three bits for exponent.

    (a) 7_{10}

    (b) 0.25_{10}

    (c) -7_{10}

    (d) -0.25_{10}
12. Determine the dead band of the system

    $y(n) = 0.2y(n-1) + 0.5y(n-2) + x(n)$

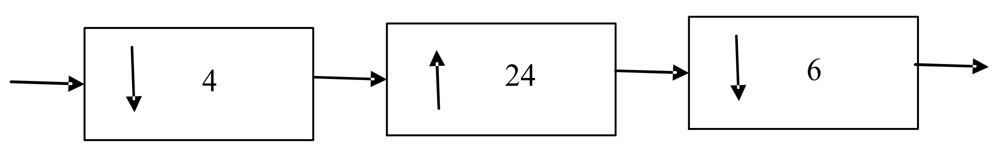
    Assume 8 bits are used for signal representation.

UNIT – V

MULTI RATE SIGNAL PROCESSING

PART – A

1. What is decimation? (Nov2010)(Nov2012)
2. What is Sub band coding? (Nov2010)
3. State the various applications of DSP. (April2011)
4. What is echo cancellation? (April2011)
5. What is multi rate signal processing? (Nov2011)
6. What is meant by down sampling and up sampling? (Nov2011)
7. Give the applications of multi rate sampling? (May2012)
8. What are called poly phase filters? (May2012)
9. Find the expression for the following multi rate system? (Nov2012)



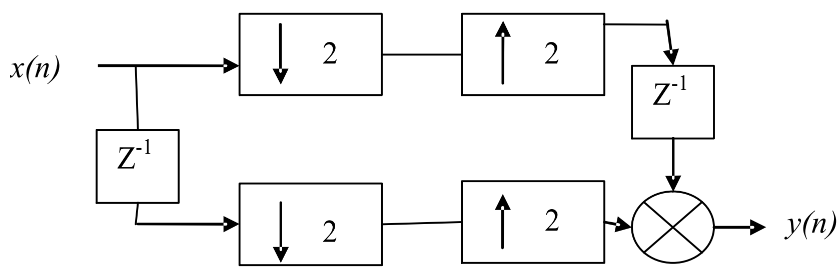
10. What is anti – imaging filter? (May2013)
11. Give the applications of multi rate DSP (May2013) (Nov2013)
12. Give the steps in multistage sampling rate converter design (Nov2013)
13. What is interpolation?
14. What is interpolator? Draw the symbolic representation of an interpolator?
15. What is anti aliasing filter?
16. What is anti – imaging filter?
17. What is poly phase decomposition?
18. What are the advantages of multi rate processing?
19. What is decimator? Draw the symbolic representation of a decimator?
20. If the spectrum of a sequence $x(n)$ is $X(e^{j\omega})$, then what is the spectrum of a signal down sampled by a factor 2?

PART – B

1. Explain the poly phase structure of decimator and interpolator?(Nov2010)
2. Discuss the procedure to implement digital filter bank using multi rate signal processing? (Nov2010)
3. (i) Explain how various sound effects can be generated with the help of DSP?

   (ii) State the applications of multirate signal processing? (May2011)
4. (i) Explain how DSP can be used for speech processing?

   (ii) Explain in detail about decimation and interpolation? (May2011)
5. For the multi rate system shown in figure, find the relation between *x(n)* and *y(n)* (Nov2011)



6. Explain the efficient transversal structure for decimator and interpolator? (Nov2011)
7. Explain sub band coding in detail (May2012)
8. Explain sampling rate conversion by a rational factor and derive input and output relation in both time and frequency domain (Nov2012)
9. Explain the design of narrow band filter using sampling rate conversion(Nov2012)
10. Explain the design steps involved in the implementation of multistage sampling rate converter. (Nov2013)
11. Explain the implementation steps in speech coding using transform coding?(Nov2013)
12. A signal *x(n)* is given by *x(n)* = {0,1,2,3,4,5,6,0,1,2,3….}

    (i) Obtain the decimated signal with a factor of 2.

    (ii) Obtain the interpolated signal with a factor of 2. (May2013)